Claims

1. Method of separating acoustic signals from a plurality of sound sources (S1, S2), comprising the following steps:

- disposing two microphones (MIK1, MIK2) at a predefined distance (d) from one another;
- picking up the acoustic signals with both microphones (MIK1, MIK2) and generating associated microphone signals (m1, m2); and
- separating the acoustic signal of one of the sound sources (S1) from the acoustic signals of the other sound sources (S2) on the basis of the microphone signals (m1, m2),

in which the separation step comprises the following steps:

- applying a Fourier transform to the microphone signals in order to determine their frequency spectra (M1, M2);
- determining the phase difference (φ) between the two microphone signals (m1, m2) for every frequency component of their frequency spectra (M1, M2);
- determining the angle of incidence (ϑ) of every acoustic signal allocated to a frequency of the frequency spectra (M1, M2) on the basis of the phase difference (φ) and the frequency;
- generating a signal spectrum (S) of a signal to be output by correlating one of the two frequency spectra (M1, M2) with a filter function (F_{ϑ_0}) which is selected so that acoustic signals from an area (γ_{3db}) around a preferred angle of incidence (ϑ_0) are amplified relative to acoustic

signals from outside this area (γ_{3db}); and

- applying an inverse Fourier transform to the resultant signal spectrum,

characterised in that the filter function (F_{ϑ_0}) is dependent on ϑ and has a maximum at the preferred angle of incidence (ϑ_0) when ϑ is varied, and the correlation of the filter function (F_{ϑ_0}) with one of the two frequency spectra comprises multiplying the same.

2. Method as claimed in claim 1, characterised in that the filter function (F_{ϑ_0}) is expressed as follows:

$$F_{\vartheta_0}(f,T)=Z(\vartheta-\vartheta_0)+D\Delta^2{}_f Z(\vartheta-\vartheta_0)$$

in which

f is the respective frequency

T is the instant at which the frequency spectra (M1, M2) are determined

$Z(\vartheta-\vartheta_0)$ is an allocation function with a maximum at ϑ_0

$D \geq 0$ is a diffusion constant and

$\Delta^2{}_f$ is a discrete diffusion operator.

3. Method as claimed in claim 2, characterised in that the allocation function (Z) is expressed as follows:

$$Z(\vartheta-\vartheta_0)=\left(\frac{1+\cos(\vartheta-\vartheta_0)}{2}\right)^n$$

where $n > 0$.

4. Method as claimed in one of claims 1 to 3, characterised in that the angle of incidence ϑ is determined by the equation

$$\vartheta=\arccos(x(f,T))$$

with

$$x(f,T)=\varphi c/2\pi fd$$

where

φ is the phase difference between the two microphone signal components (m1, m2)

c is the acoustic velocity

f is the frequency of the acoustic signal component and

d is the predefined distance of the two microphones (MIK1, MIK2).

5. Method as claimed in claim 4, characterised in that it additionally incorporates the following step:

limiting the value of x(f,T) to the interval [-1,1].

6. Method as claimed in claim 5, characterised in that it additionally incorporates the following step:

reducing signal components whose value of x(f,T) lay outside of the interval [-1,1] prior to limitation.

7. Device for implementing the method as claimed in one of claims 1 to 6, comprising:

- two microphones (MIK1, MIK2);
- a sampling and Fourier transform unit (20) connected to the microphones for discretizing and digitising the microphone signals (m1, m2) and applying a Fourier transform to them;
- a calculating unit (30) connected to the sampling and Fourier transform unit (20) for calculating the angle of incidence (ϑ) of every acoustic signal component; and
- at least one signal generator (40) connected to the calculating unit (30) for outputting the

separated acoustic signal, at least one signal generator (40) having means for multiplying one of the Fourier transformed frequency spectra (M1, M2) by a filter function (F_{ϑ_0}) which is dependent on ϑ and has a maximum at a preferred angle of incidence (ϑ_0) when ϑ is varied.

8. Device as claimed in claim 7, characterised in that the distance (d) between the microphones satisfies the equation:

$$d<c/4f_A$$

where c is the acoustic velocity and f_A is the sampling frequency of the stereo sampling and Fourier transform unit (20).

9. Device as claimed in claim 7 or 8, characterised in that the device has a signal generator (40) for every sound source (S1, S2) to be separated.